The Real time Transport Protocol (RTP) is an Internet protocol standard that defines a way for applications to manage the real-time transmission of multimedia data. RTP is used at the bearer or media level (as opposed to the call control level which employs SIP or other call control protocol) for Internet telephony applications including VoIP. RTP does not guarantee real-time delivery of multimedia data, as this is dependent on the actual network characteristics. RTP provides the functionality to manage the data as it arrives to best effect. User Plane Adaptation (UPA) is the procedure used by the MRF and a given UE to monitor the RTP traffic between them and to adjust bandwidth utilisation in an attempt to provide optimal quality during a talk session. UPA provides for the MRF to dynamically redefine the talk burst duration which is encapsulated in a given RTP packet on a given link (this parameter is known as *ptime*) and the codec used for that link (the codec is identified by one of a number of parameters contained in a "mode set"). The SIP message reINVITE/UPDATE is used to signal these parameters to the UE. The UE may also send this message to the MRF in order to notify the MRF of its capabilities/requirements.

The group know as the Open Mobile Alliance has developed a Push to talk Over Cellular (PoC) specification aimed at enabling the provision of services over standard mobile networks which resemble walkie-talkie services, i.e. at the push of a button a subscriber can be instantly connected to one or more other subscribers. PoC relies upon the MRF to set up and handle connections. The PoC specification describes the tools available to detect packet loss over the links between the MRF and individual UEs. PoC also describes a means to request a change in bandwidth utilization, but does not provide detailed algorithms or procedures to enable this.

Summary of the Invention

According to a first aspect of the present invention there is provided a method of optimising the bandwidth usage on a Real-Time Protocol managed link transporting media from a Media Resource Function of a cellular telecommunications network to User Equipment, the method comprising:

monitoring the rate of packet loss of the link to determine whether the rate of packet loss is unacceptably high or within acceptable limits; and

as a result of said monitoring, adapting the sending rate over the link by re-packetising media, received at the Media Resource Function from third party nodes, to either increase the size of packets sent over the link when the rate of packet loss is unacceptably high, thereby reducing packet header overhead and reducing bandwidth usage on the link; or to decrease the size of packets sent over the link when the rate of packet loss is within acceptable limits, thereby reducing the transmission delay over the link.

The invention is applicable in particular to networks in which the Media Resource Function is arranged to handle media distribution for Push-to-talk over Cellular services.

Embodiments of the present invention have the advantage that adaption on the downlink can be achieved without having to vary the packet sizes transmitted by third party nodes. Thus, transmission delays on these uplinks to the Media Resource Function are maintained at optimum levels. An additional consequential benefit is that bandwidth usage can be adapted without having to signal to other UEs. Expensive additional signalling traffic is thus avoided.

Preferably, the method comprises re-packetising received media only into packet sizes which are larger than the packet sizes in which the media is received at the Media Resource Function.

Typically, the step of monitoring the rate of packet loss of the link comprises sampling the rate of packet loss on the link. This may be carried out at the receiving UE, with the UE sending the samples to the Media Resource Function. The Media Resource Function adjusts the sent packet size in order to reduce the rate of packet loss on the link or to decrease the transmission delay. In particular, when the rate of packet loss is unacceptably high, the Media Resource Function may re-packetise incoming media into larger packets, thereby reducing the packet header overhead and reducing the bandwidth usage on the downlink. When the rate of packet loss is within acceptable limits, the

incoming media may be re-packetised to reduce the packet size, thereby reducing the transmission delay over the link.

It will be appreciated that said step of adapting the sending rate is carried out dynamically in response to the monitored rate of packet loss.

Preferably, in the event that media is to be repacketised at the Media Resource Function, received media is stored at the Media Resource Function in a buffer until such time as sufficient media has been received to construct a packet of the necessary size.

Said third party nodes are typically peer User Equipment (UEs), although they may be other nodes such as web servers, etc.

According to a second aspect of the present invention there is provided a Media Resource Function node for use in a cellular telecommunications network, the node handling media sent between itself and user equipment over a Real-Time Protocol managed link, the node comprising:

means for monitoring the rate packet loss of the downlink to the User Equipment to determine whether the rate of packet loss is unacceptably high or within acceptable limits; and

means for adapting, based upon the monitored packet loss, the sending rate over the link by re-packetising media received from third party nodes, to increase the size of packets sent over said downlink when the rate of packet loss is unacceptably high, thereby reducing packet header overhead and reducing bandwidth usage on the link, or to decrease the size of packets sent over the link when the rate of packet loss is within acceptable limits, thereby reducing the transmission delay over the link.

Brief Description of the Drawings

Figure 1 illustrates schematically the architecture of a cellular telecommunications network employing a MRF node to coordinate VoIP voice conferencing; and
Figure 2 is a flow diagram illustrating a method for adapting bandwidth usage on a link of a VoIP voice conference.

Detailed Description of Certain Embodiments

Considering now in detail the PoC service, a single MRF can manage thousands of talk sessions, with each session being independent from other sessions that may be hosted by the same MRF. A given talk session will comprise two or more pieces of user equipment (UE) and the central MRF. These UEs might each have different capabilities. Talk bursts from a UE are encoded and sent to the MRF (via respective GGSNs) as one or more RTP packets, referred to here as simply "packets". The MRF then forwards the packets to the or each other UE participating in the same talk session. The path which the packets take from the UE to the MRF is called the "uplink". The

Claims

1. A method of optimising the bandwidth usage on a Real-Time Protocol managed link transporting media from a Media Resource Function of a cellular telecommunications network to User Equipment, the method comprising:

monitoring the rate of packet loss of the link to determine whether the rate of packet loss is unacceptably high or within acceptable limits; and

as a result of said monitoring, adapting the sending rate over the link by re-packetising media, received at the Media Resource Function from third party nodes, to either increase the size of packets sent over the link when the rate of packet loss is unacceptably high, thereby reducing packet header overhead and reducing bandwidth usage on the link; or to decrease the size of packets sent over the link when the rate of packet loss is within acceptable limits, thereby reducing the transmission delay over the link.

2. A method according to claim 1, wherein the step of monitoring the rate of packet loss of the link comprises sampling.

3. A method according to claim 1 or 2, wherein said step of adapting the sending rate is carried out dynamically in response to the monitored rate of packet loss.

4. A method according to any one of the preceding claims, wherein, in the event that media is to be repacketised at the Media Resource Function, received media is stored at the Media Resource Function in a buffer until such time as sufficient media has been received to construct a packet of the necessary size.

5. A method according to any one of the preceding claims, wherein said third party nodes are peer User Equipment (UEs).

6. A Media Resource Function node for use in a cellular telecommunications network, the node handling media sent between itself and user equipment over a Real-Time Protocol managed link, the node comprising:

means for monitoring the rate of packet loss of the downlink to the User Equipment to determine whether the rate of packet loss is unacceptably high or within acceptable limits; and

means for adapting, based upon the monitored properties, the sending rate over the link by re-packetising media received from third party nodes, to increase the size of packets sent over said downlink when the rate of packet loss is unacceptably high, thereby reducing packet header overhead and reducing bandwidth usage on the link; or to decrease the size of packets sent over the link when the rate of packet loss is within acceptable limits, thereby reducing the transmission delay over the link.